

ECE 440 - Introduction to Computer Networks

Spring 2020

TCP fast retransmit

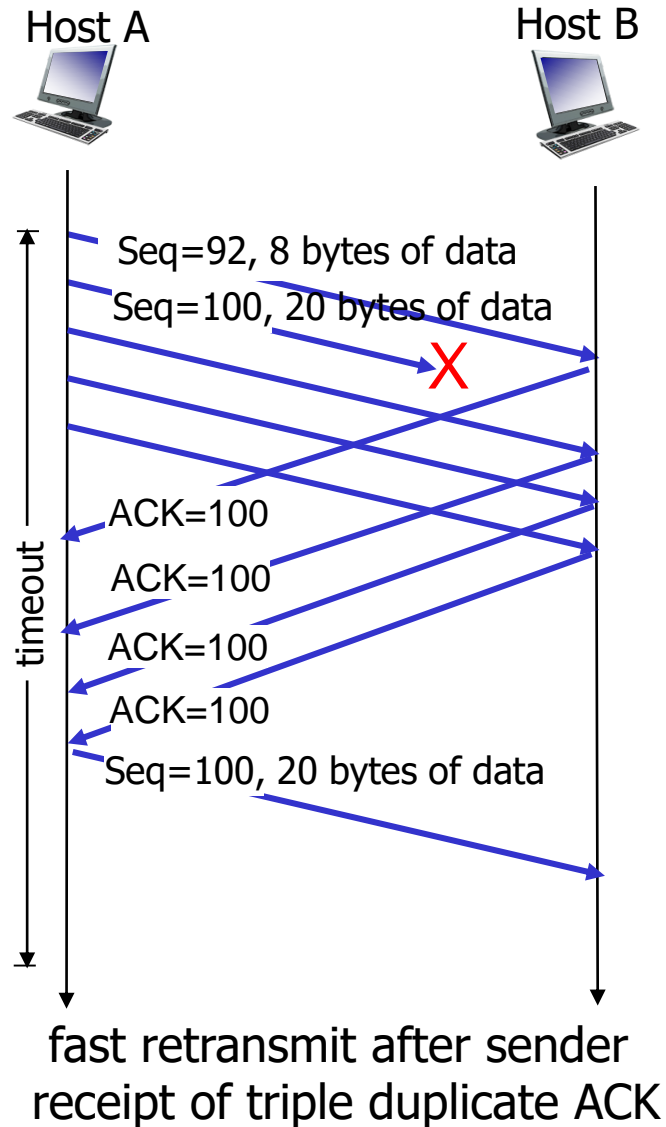
- time-out period often relatively long:
 - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
 - sender often sends many segments back-to-back
 - if segment is lost, there will likely be many duplicate ACKs.

TCP fast retransmit

if sender receives 3 ACKs for same data (“triple duplicate ACKs”), resend unacked segment with smallest seq #

- likely that unacked segment lost, so don't wait for timeout

TCP fast retransmit



Chapter 3 outline

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

3.4 principles of reliable data transfer

3.5 connection-oriented transport: TCP

- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

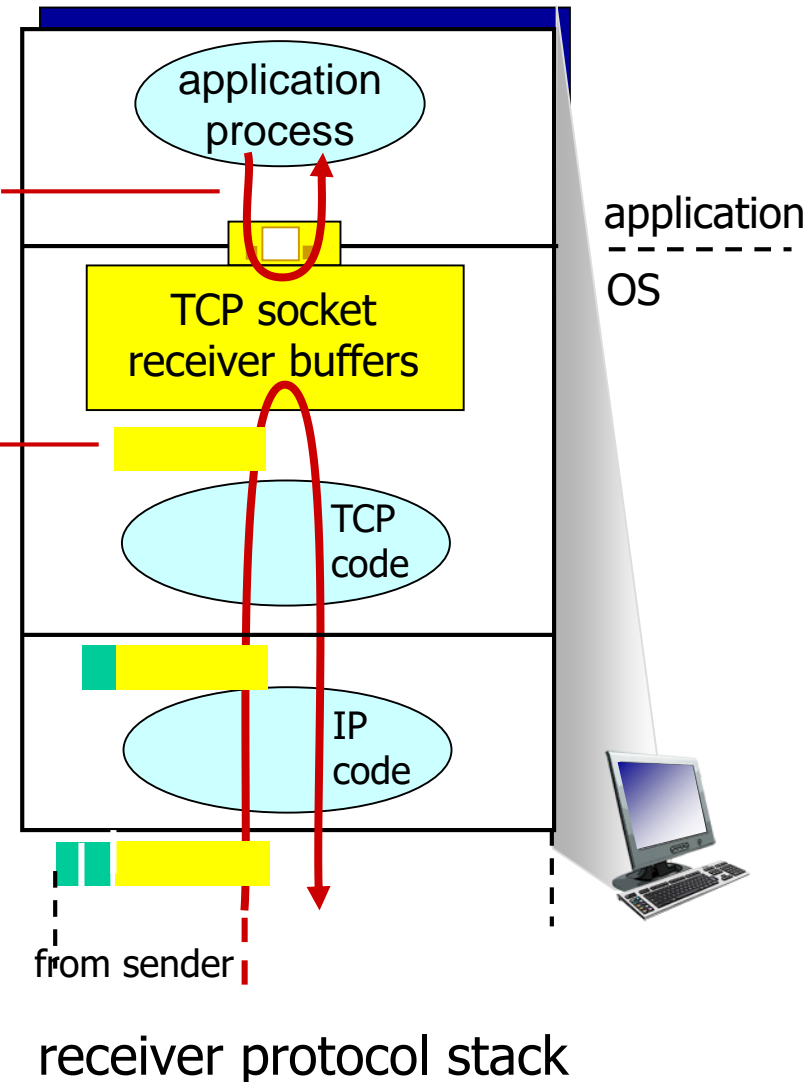
3.7 TCP congestion control

TCP flow control

application may
remove data from
TCP socket buffers

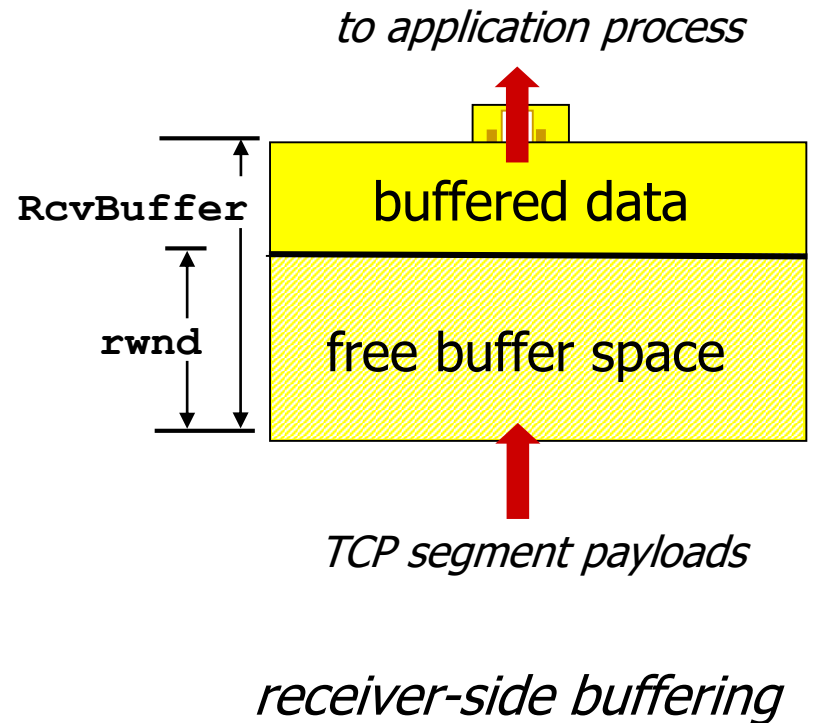
... slower than TCP
receiver is delivering
(sender is sending)

flow control
receiver controls sender, so
sender won't overflow
receiver's buffer by transmitting
too much, too fast



TCP flow control

- receiver “advertises” free buffer space by including **rwnd** value in TCP header of receiver-to-sender segments
 - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
 - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unacked (“in-flight”) data to receiver’s **rwnd** value
- guarantees receive buffer will not overflow



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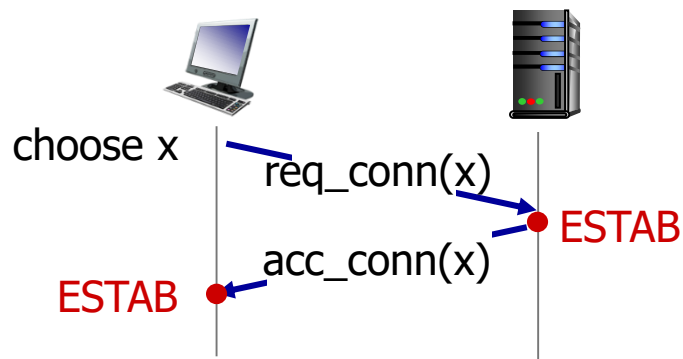
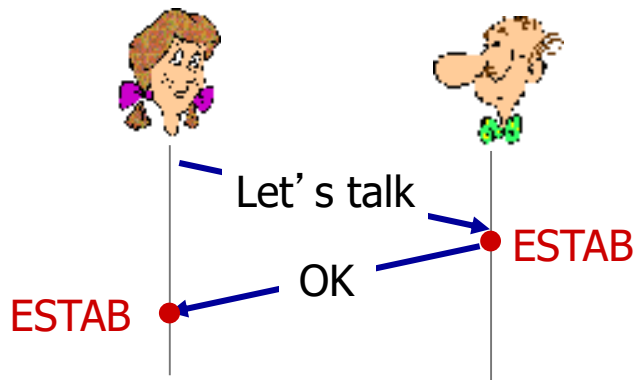
Connection Management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters

Agreeing to establish a connection

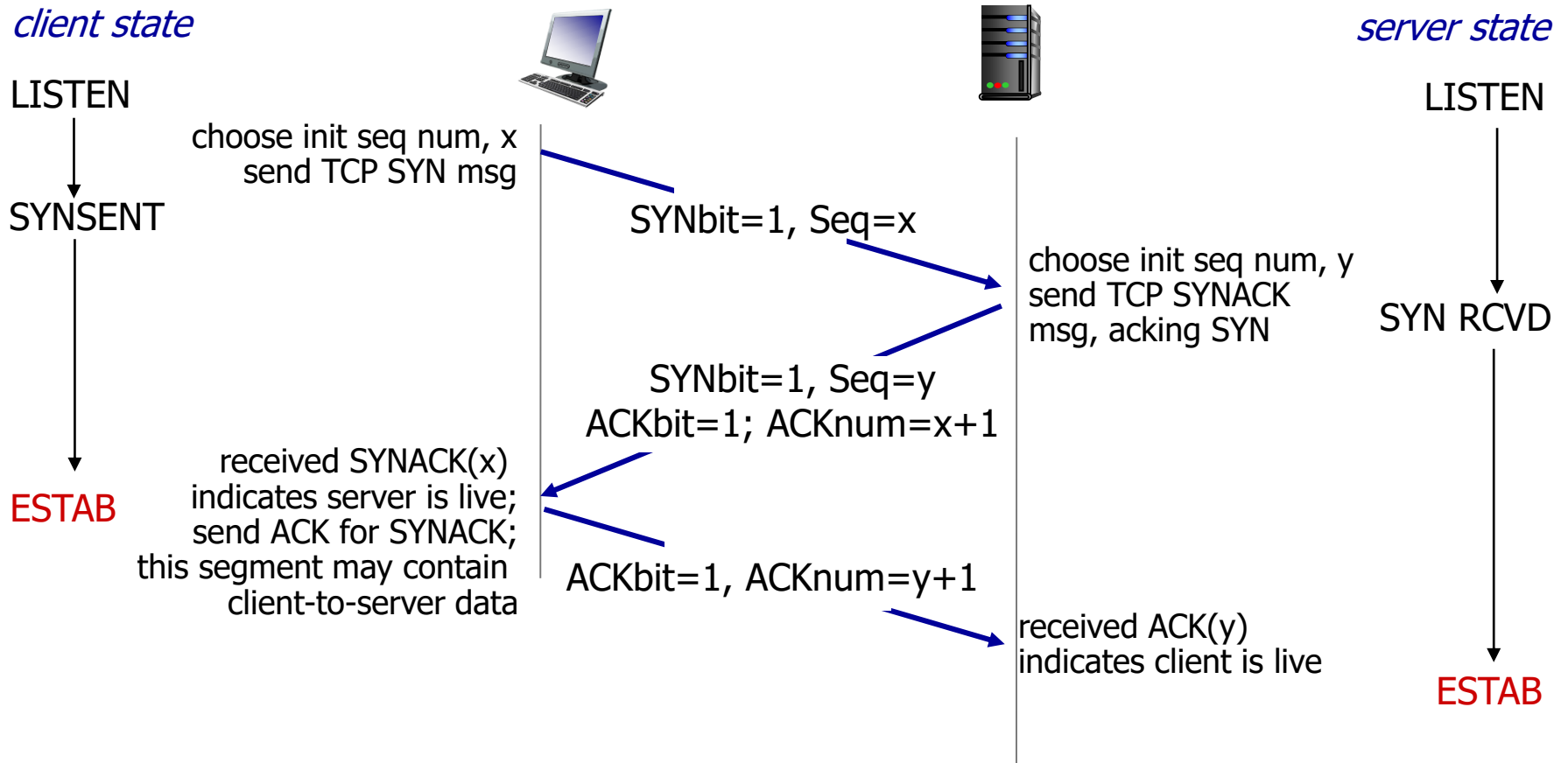
2-way handshake:



Q: will 2-way handshake always work in network?

- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

TCP 3-way handshake



TCP: closing a connection

- client, server each close their side of connection
 - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
 - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

TCP: closing a connection

client state

ESTAB

`clientSocket.close()`

FIN_WAIT_1

can no longer
send but can
receive data

FIN_WAIT_2

wait for server
close

TIMED_WAIT

timed wait
for $2 * \text{max}$
segment lifetime

CLOSED



FINbit=1, seq=x

ACKbit=1; ACKnum=x+1

FINbit=1, seq=y

ACKbit=1; ACKnum=y+1

can still
send data

can no longer
send data

server state

ESTAB

CLOSE_WAIT

LAST_ACK

CLOSED

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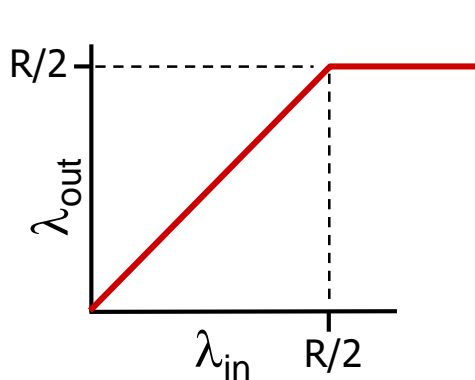
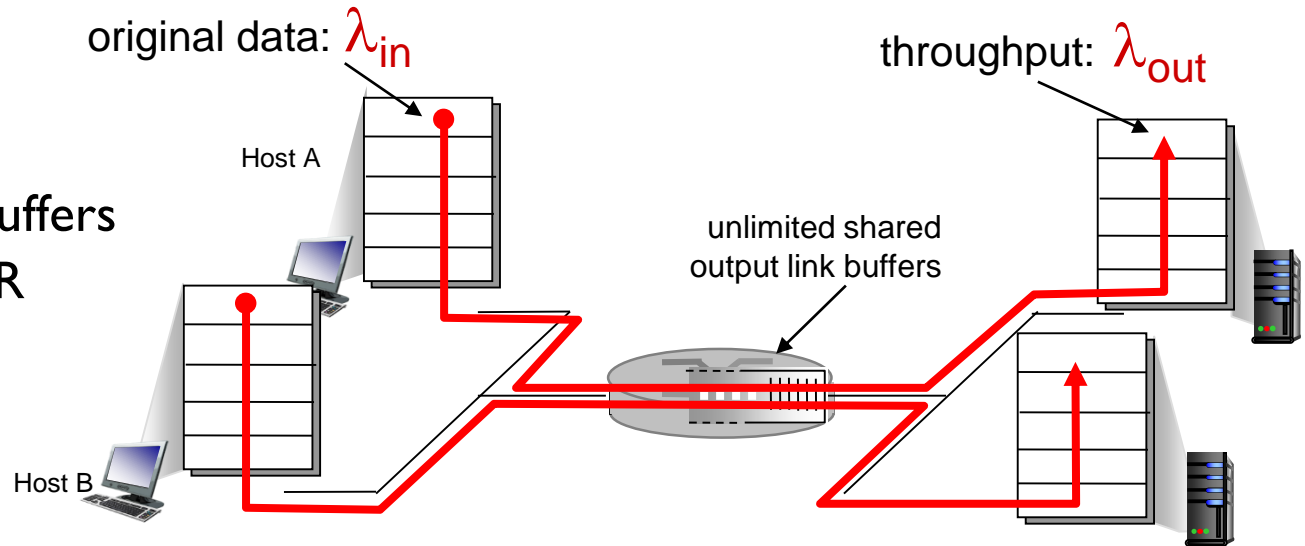
Principles of congestion control

congestion:

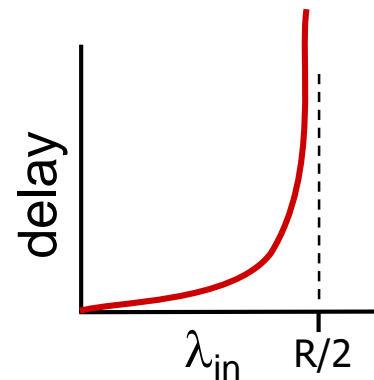
- informally: “too many sources sending too much data too fast for *network* to handle”
- different from flow control!
- manifestations:
 - lost packets (buffer overflow at routers)
 - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario I

- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission



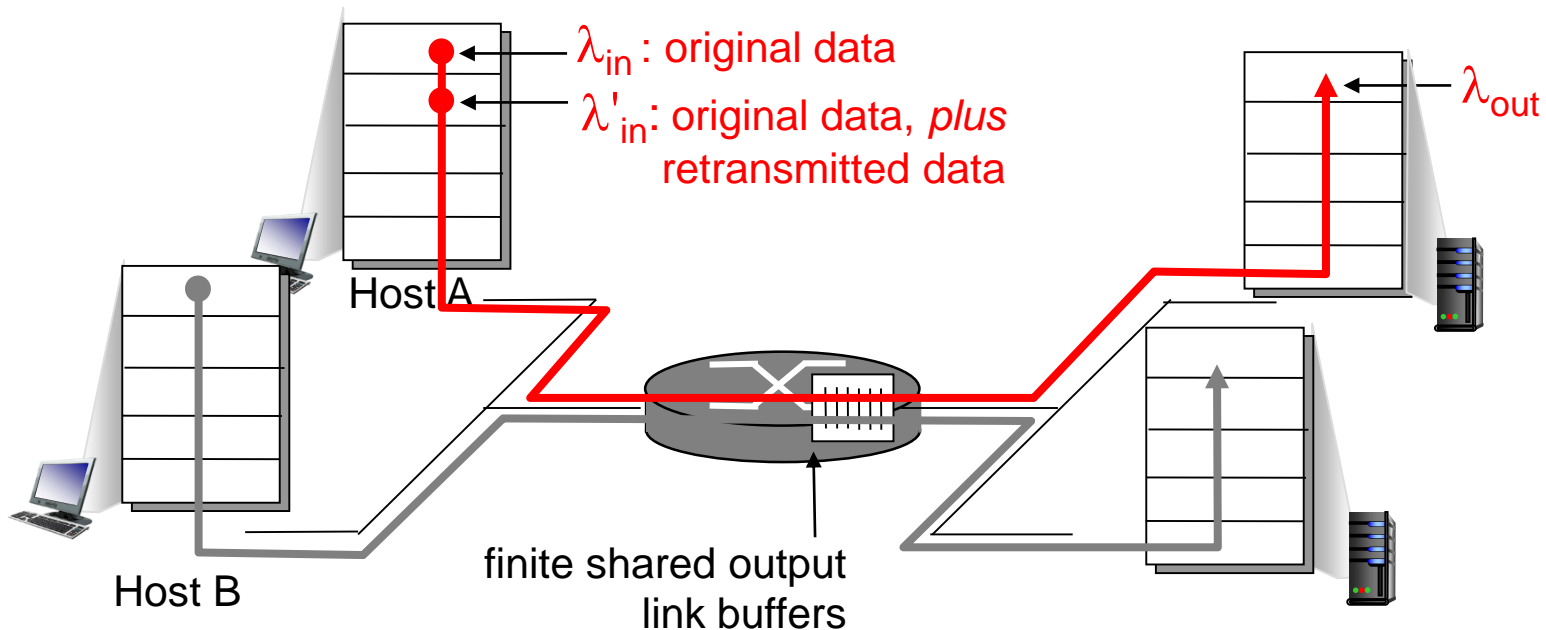
- maximum per-connection throughput: $R/2$



- ❖ large delays as arrival rate, λ_{in} , approaches capacity

Causes/costs of congestion: scenario 2

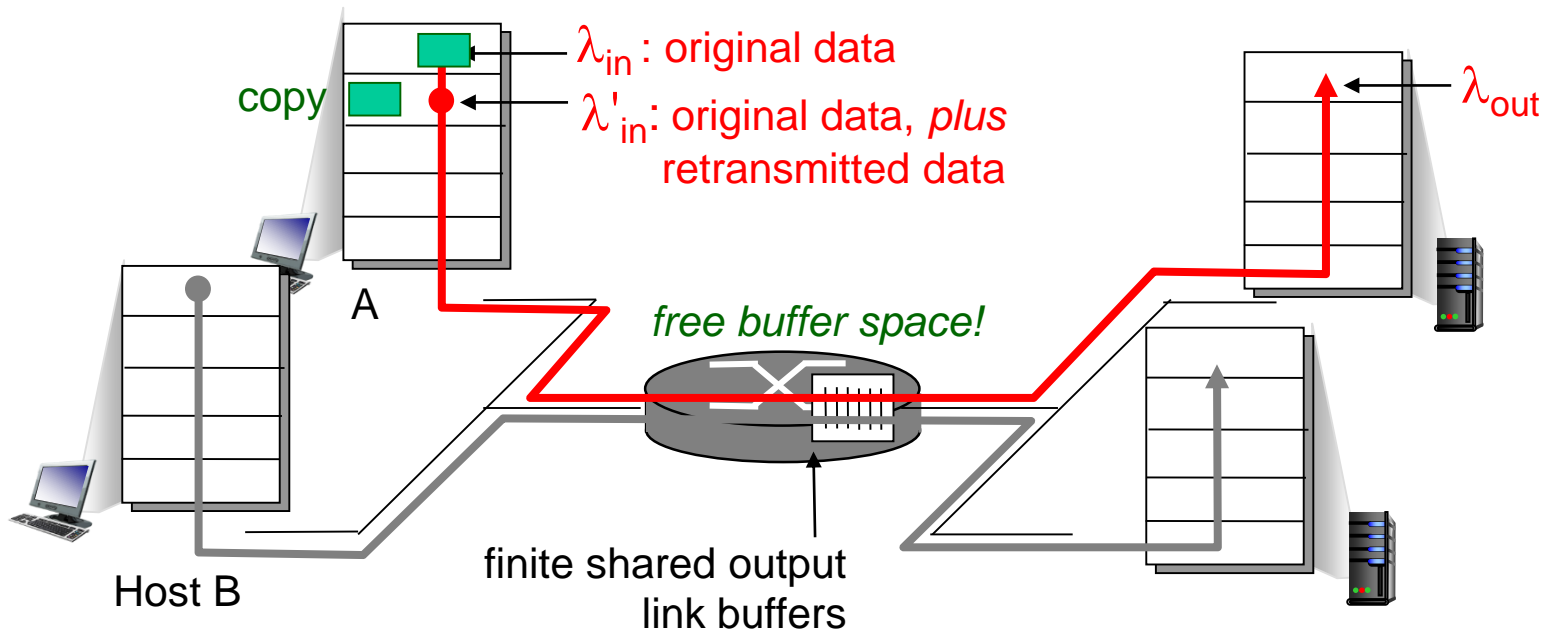
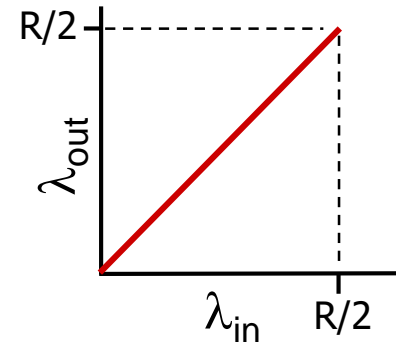
- one router, *finite* buffers
- sender retransmission of timed-out packet
 - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
 - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$



Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available

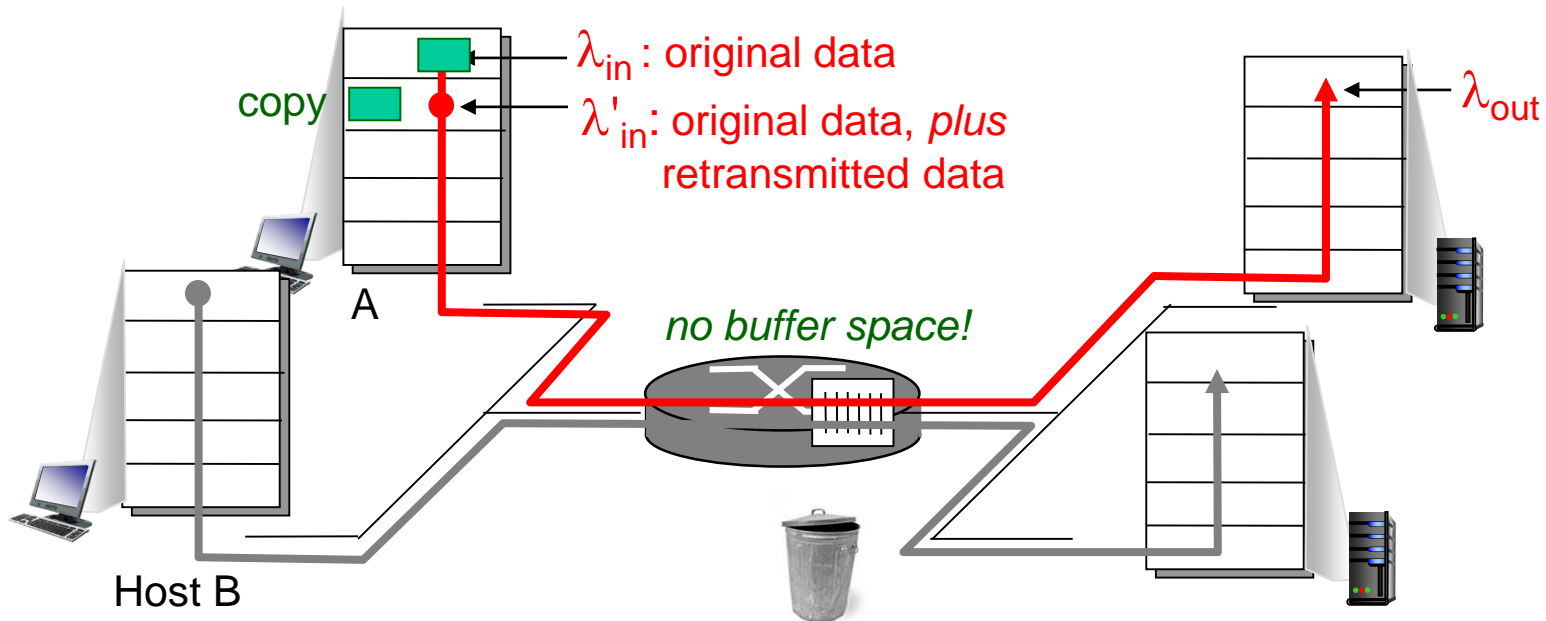


Causes/costs of congestion: scenario 2

Idealization: known loss

packets can be lost,
dropped at router due
to full buffers

- sender only resends if packet *known* to be lost

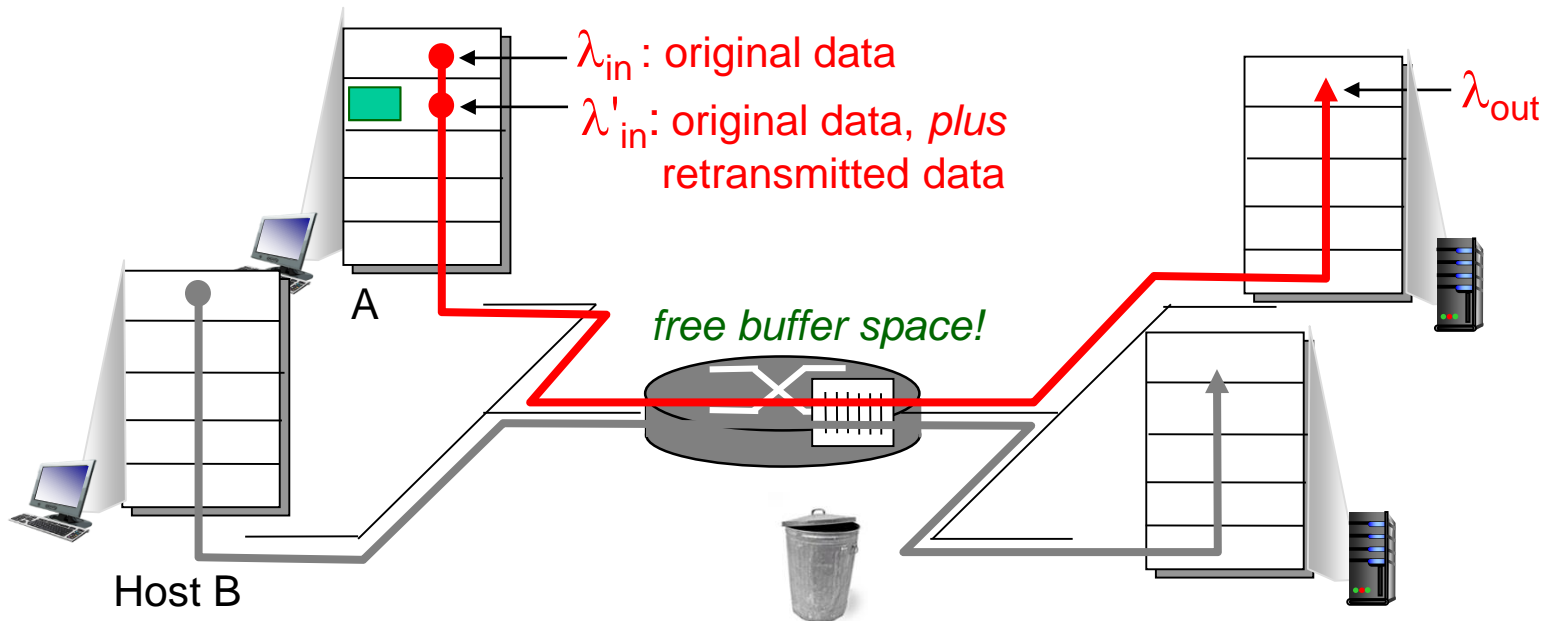
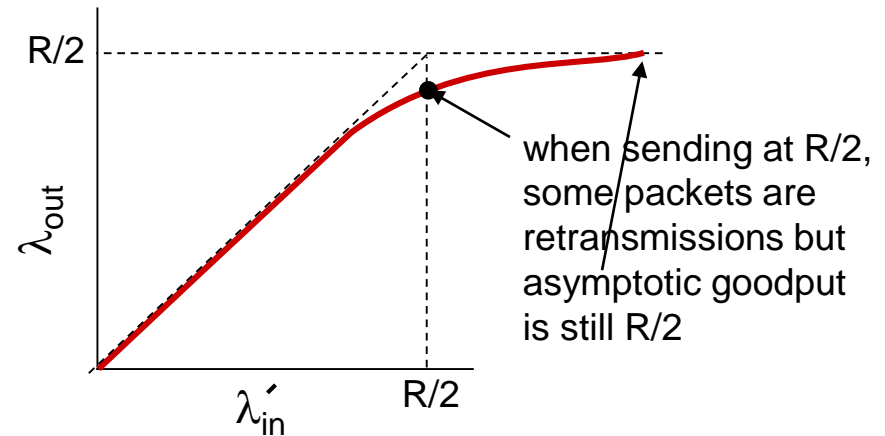


Causes/costs of congestion: scenario 2

Idealization: known loss

packets can be lost,
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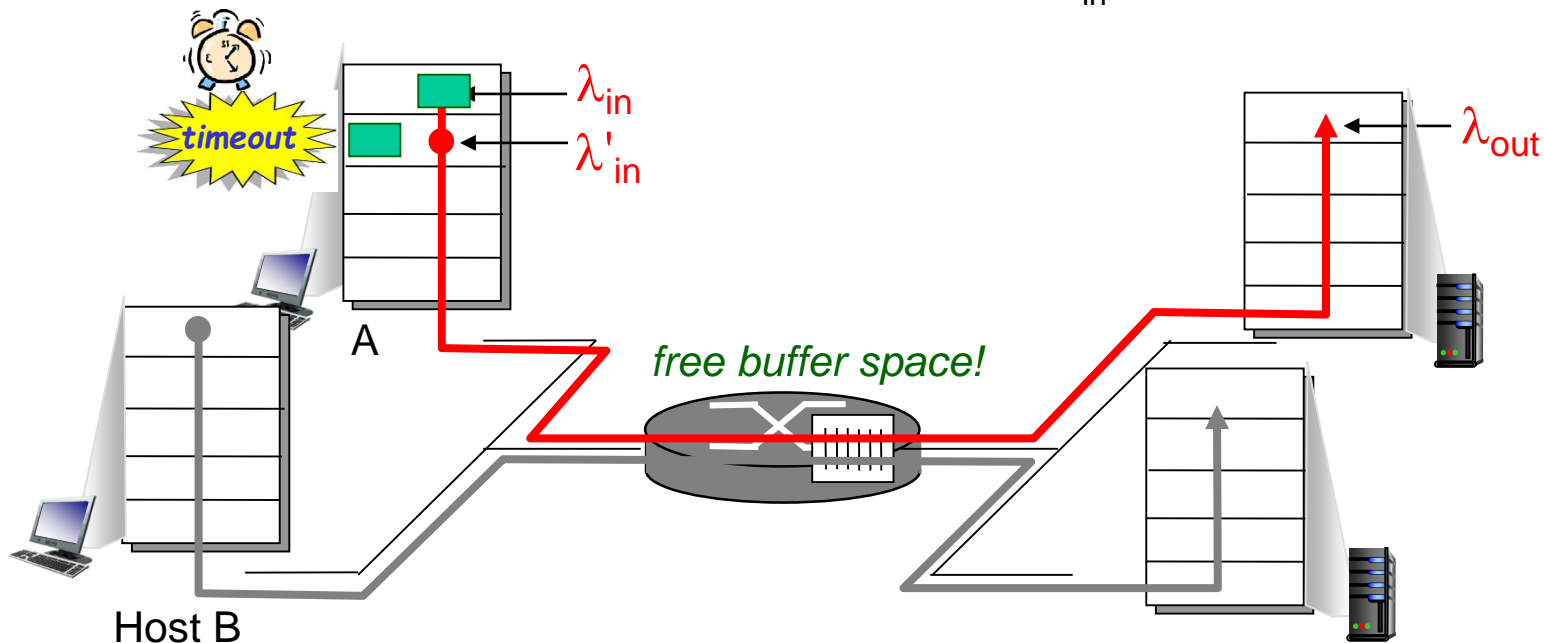
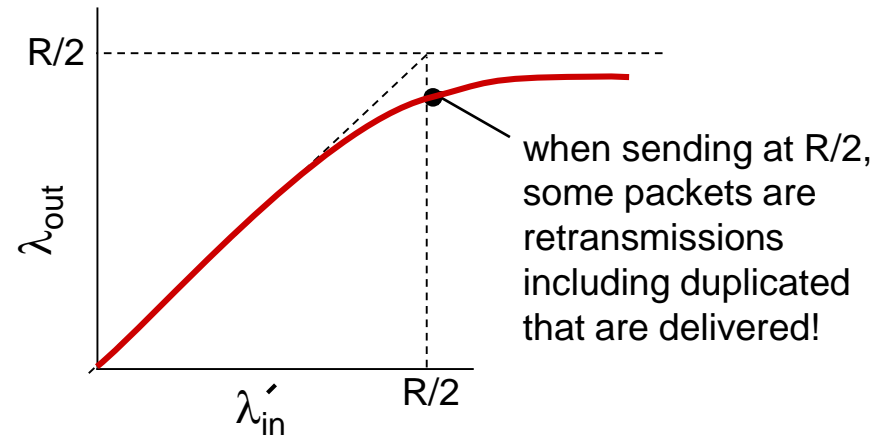
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Causes/costs of congestion: scenario 2

Realistic: *duplicates*

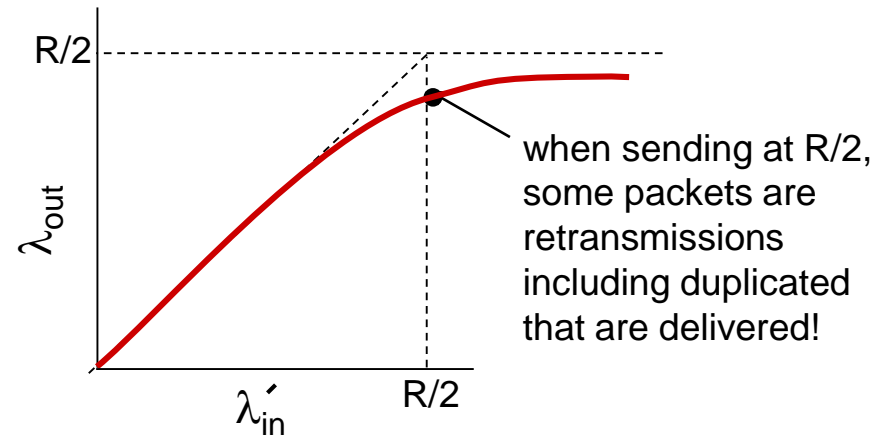
- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered



Causes/costs of congestion: scenario 2

Realistic: *duplicates*

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered



“costs” of congestion:

- more work (retrans) for given “goodput”
- unneeded retransmissions: link carries multiple copies of pkt
 - decreasing goodput

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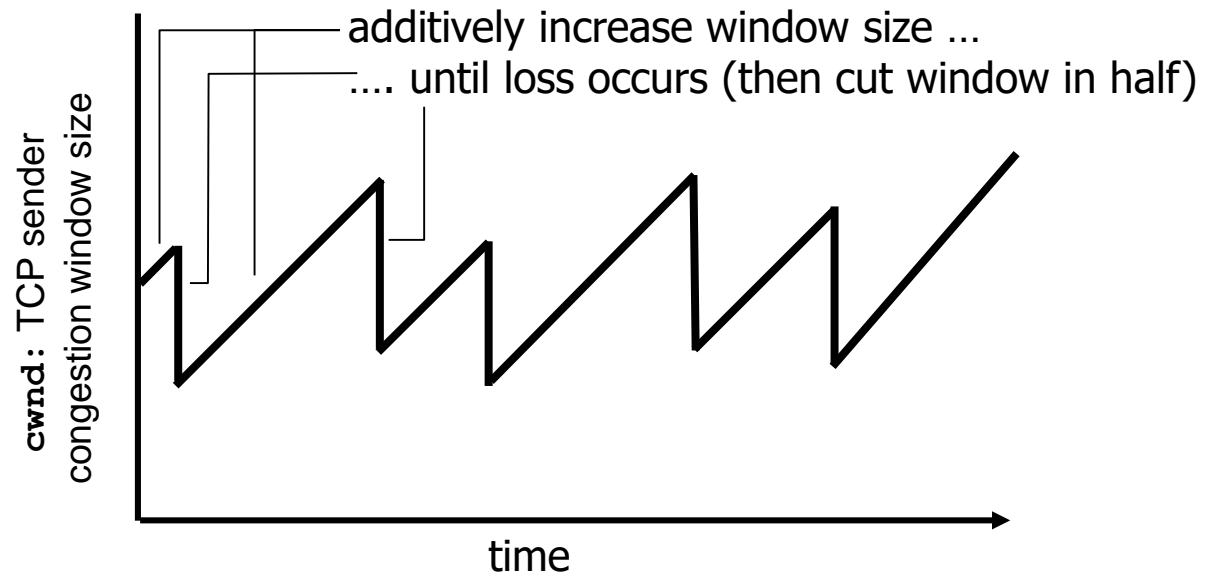
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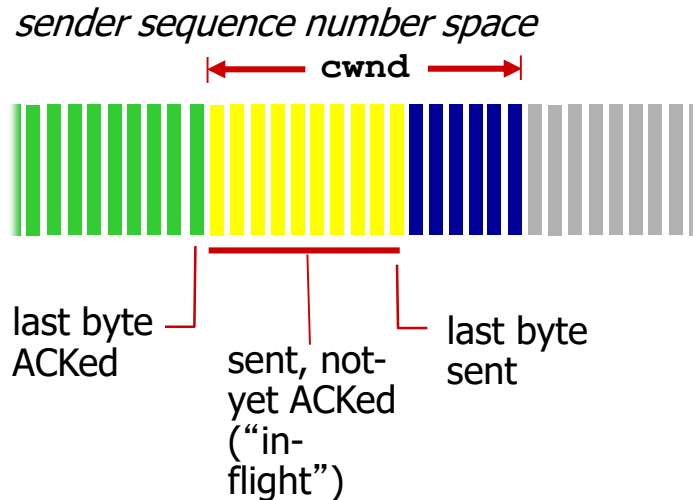
TCP congestion control: additive increase multiplicative decrease

- *approach*: sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
 - *additive increase*: increase **cwnd** by 1 MSS every RTT until loss detected
 - *multiplicative decrease*: cut **cwnd** in half after loss

AIMD saw tooth
behavior: probing
for bandwidth



TCP Congestion Control: details



- sender limits transmission:

$$\text{LastByteSent} - \text{LastByteAcked} \leq \text{cwnd}$$

- **cwnd** is dynamic, function of perceived network congestion

TCP sending rate:

- *roughly*: send cwnd bytes, wait RTT for ACKS, then send more bytes

$$\text{rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$